MultiVOIP™ 100

Voice/FaxoverIPNetworks

Standalone Voice/IP Gateway

Model MVP120
H.323 Mode

Quick Start Guide

MultiTech® Systems

Quick Start Guide

MultiVOIP 100 (Model MVP120)
PN 82060450 Revision A

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Patents

This Product is covered by one or more of the following U.S. Patent Numbers: ***5.682.386***; ***5.757.801; 6.151.333***. Other Patents Pending.

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Technical Support

Multi-Tech Systems has an excellent staff of technical support personnel available to help you get the most out of your Multi-Tech product. If you have any questions about the operation of this unit, or experience difficulty during installation you can contact Tech Support:

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Introduction

The MultiVOIP model MVP120 allows analog voice and fax communication over an IP network. Multi-Tech's voice/fax gateway technology allows voice/fax communication to be transmitted, with no additional expense, over your existing IP network, which has traditionally been data-only. To access this free voice and fax communication, all you have to do is connect the MVP120 to your telephone equipment, and then to your existing Internet connection. Once configured, the MVP120 then allows voice and fax to travel down the same path as your traditional data communications.

The MVP120 supports the H.323 standards-based protocol enabling your MVP120 to communicate with other third-party VOIP Gateways or other endpoints that support the H.323 protocol, such as Microsoft NetMeeting®. The H.323 standard defines how endpoints make and receive calls, how endpoints negotiate a common set of audio and data capabilities, and how information is formatted and sent over the network. This version of the software also supports communication with a Multi-Tech MVPGK1 Gatekeeper or an optional 3rd party H.323 Gatekeeper which, when enabled, maintains its own phonebook database, pre-registers all endpoints, controls the bandwidth, and handles all conferencing issues such as transferring of calls.

Figure 1. MVP120

The MVP120 has one FXO voice/fax channel, a 10 Mbps Ethernet LAN interface, and a Command port for configuration.

System management is provided through the Command port using bundled Windows® software which provides easy-to-use configuration menus and comprehensive online Help.

Related Documentation

The MVP120 Quick Start Guide is intended to be used by qualified systems administrators and network managers. This Quick Start Guide provides the necessary information for a qualified person to unpack, cable, load software, and configure the unit for proper operation.

A detailed MVP120 User Guide is included on your system CD and provides in-depth information on the features and functionality of Multi-Tech's MVP120.

The CD media is produced using Adobe Acrobat™ for viewing and printing the user guide. To view or print your copy of a user guide, load Acrobat Reader™ onto your system from the CD, or obtain it as a free download from Adobe's Web site:

http://www.adobe.com

The MVP120 User Guide is also available from Multi-Tech's Web site:

http://www.multitech.com

Installing Your MVP120

This section covers the basic installation steps which include unpacking the MultiVOIP, connecting the cables, and configuring the MVP120.

Overview of the Installation and Configuration Process

The VOIP administrator must first install the MVP120 software and then configure each MVP120 for its specific function. During the configuration process, it's important to note that the Phone Directory Database is configured differently depending on whether or not you have Gatekeeper support on your VOIP network.

If your VOIP network supports an H.323 Gatekeeper, you must register all H.323 endpoints with the Gatekeeper. The procedure for doing this is explained in the section "Registering with a Gatekeeper Phone Directory."

If your VOIP network does not have Gatekeeper software or the Gatekeeper software is not enabled, then you must build a proprietary phonebook with a "Master" MultiVOIP and "Slave" MultiVOIPs. The "Master" unit includes the assignment of a unique LAN IP address, subnet mask, and Gateway IP address. Once all connections have been made, the VOIP administrator configures the unit and builds the Phone Directory Database that will reside in the Master unit.

After the "Master" MultiVOIP is configured, the administrator moves on to configure the MultiVOIPs designated as "Slave" units. Again, unique LAN IP addresses, subnet masks, and Gateway IP addresses are assigned. When this is done, the Phone Directory Database option is set to Slave, and the IP address of the Master MultiVOIP is entered. Once all Slave units are configured, the process moves on to the "Deploying the VOIP Network" section.

Deploying the VOIP Network

The final phase of the installation is deployment of the network. When the remote MultiVOIPs are sent to their remote sites, the remote site administrators need only connect the units to their LAN and telephone equipment. A full Phone Directory Database Proprietary Phonebook, supplied by the Master MultiVOIP will be loaded into their units within minutes of being connected and powered up.

For remote VOIPs that were configured with the **Gatekeeper** option enabled, each MultiVOIP will be registered with the Gatekeeper. The Gatekeeper phonebook directory is NOT downloaded to the remote units.

Unpacking Your MultiVOIP

Remove all items from the box. (See Figure 2.)

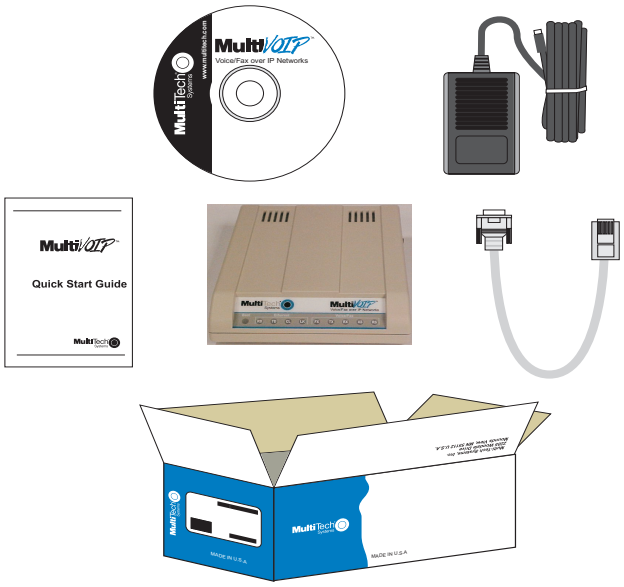


Figure 2. Unpacking

Safety Warnings

Caution: Danger of explosion if battery is incorrectly replaced.

A lithium battery on the circuit board provides backup power for the time keeping capability. The battery has an estimated life expectancy of ten years.

When the battery starts to weaken, the date and time may be incorrect. If the battery fails, the board must be sent back to Multi-Tech Systems for battery replacement.

The Ethernet port is not designed to be connected to a Public Telecommunication Network. FXO can be connected to a public or PSTN network.

Cabling Your MultiVOIP

Cabling your MultiVOIP involves properly connecting the Power, Command Port, phone/PBX, and Ethernet. Figure 3 shows the back panel connectors and the associated cable connections.

Cabling Procedure

1. Using the supplied cable, connect the power supply to a live AC outlet, then plug the power supply into the MVP120.

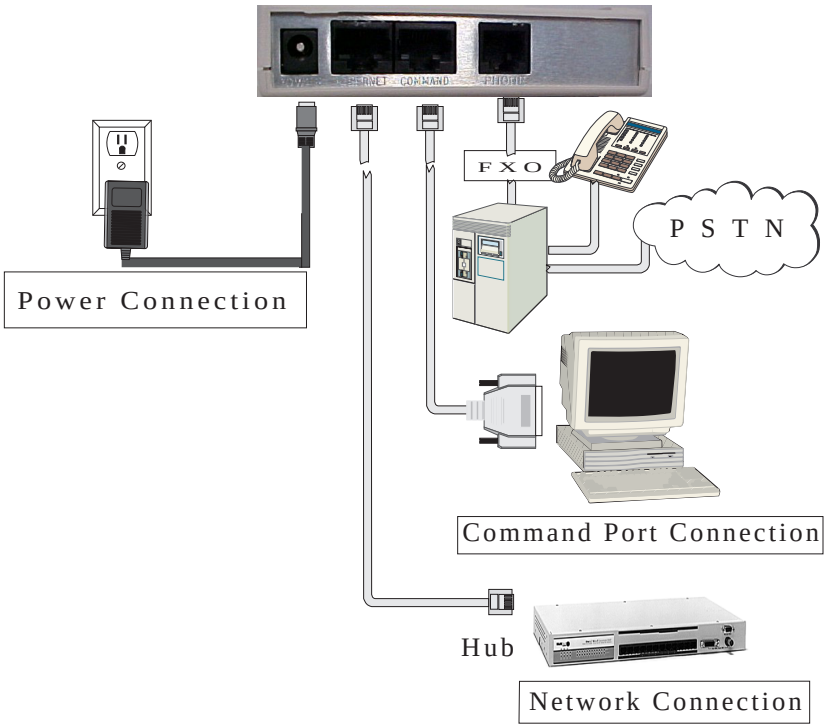


Figure 3. Cable Connections

2. Connect the MVP120 to a PC using the RJ-45 to DB9 (female) cable provided with your unit. Plug the RJ-45 end of the cable into the **Command** port of the MVP120 and connect the other end to the PC's serial port (Figure 3).

3. Connect a network cable to the **Ethernet 10Base-T** connector on the back of the MVP120. Connect the other end of the cable to your network.

4. Connect an RJ-11 cable to the **Voice/Fax Channel** connector on the back of the MVP120 and the other end to the PSTN or FXO PBX port.

5. Turn on power to the MVP120 by setting the power switch on the right side to the On position. Wait for the Boot LED on the MVP120 to go off before proceeding. This may take a couple of minutes.

   Note: Since the MVP120 doesn't have a power LED, no LEDs will be lit if there is no activity after boot up, unless the ethernet port is connected to the network.

Installing Your MVP120 Software

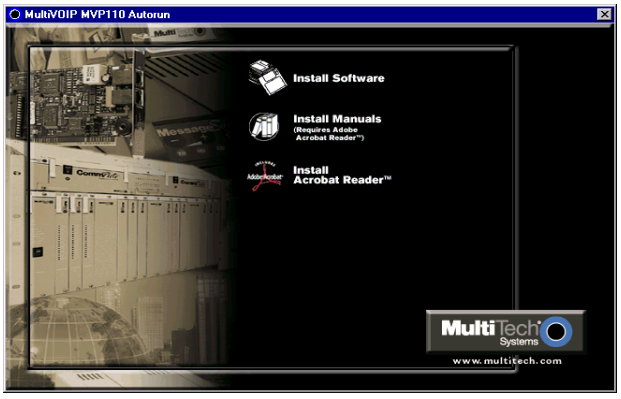
These following installation procedures do not cover every screen or option; it is assumed that a technical person with a thorough knowledge of Windows and the software loading process is doing the installation. Additional information on the MVP120 software is provided in the online Help.

Note: The phonebook directory configuration process is different depending on whether or not you have an enabled H.323 Gatekeeper resident in your network. The “Configuring Your MultiVOIP” section explains these differences.

The MVP120 software and User Guide are contained on the MVP120 Installation CD. When you have finished configuring your MVP120, you can view and print the User Guide by clicking the **Install Manuals** icon.

CAUTION: If you are installing the MVP120 behind a firewall, the Firewall must support H.323. Refer to your Firewall user documentation to enable H.323 support.

1. Make certain that your MVP120 has been properly cabled and that its power is turned on.
2. Insert the MVP120 Installation CD into a CD-ROM drive. The CD is auto-detectable, so it starts automatically. It may take 10 to 20 seconds for the Multi-Tech Installation CD window to appear.

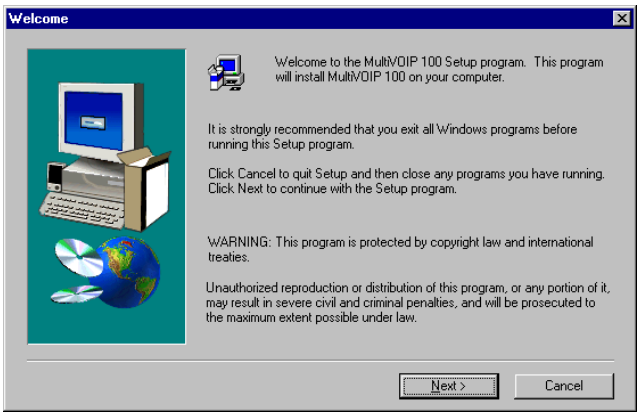


If the Multi-Tech Installation CD window does not appear automatically, click **My Computer**, then right-click the **CD-ROM drive icon**, click **Open**, then click the **Autorun** icon.

3. When the Multi-Tech Installation CD window displays, click the **Install Software** icon.

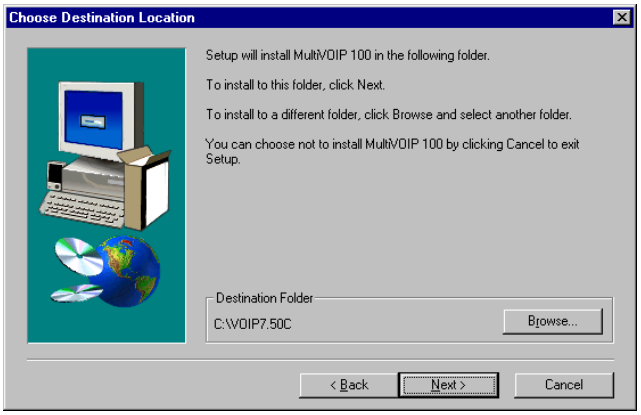
4. You will then be prompted to select your software: **H.323** or **Proprietary**.
   Select **H.323**.

5. The **Welcome/Setup** dialog box displays.


Welcome
Welcome to the MultiVOIP 100 Setup program. This program will install MultiVOIP 100 on your computer.
It is strongly recommended that you exit all Windows programs before running this Setup program.
Click Cancel to quit Setup and then close any programs you have running. Click Next to continue with the Setup program.
WARNING: This program is protected by copyright law and international treaties.
Unauthorized reproduction or distribution of this program, or any portion of it, may result in severe civil and criminal penalties, and will be prosecuted to the maximum extent possible under law.
Next >
Cancel

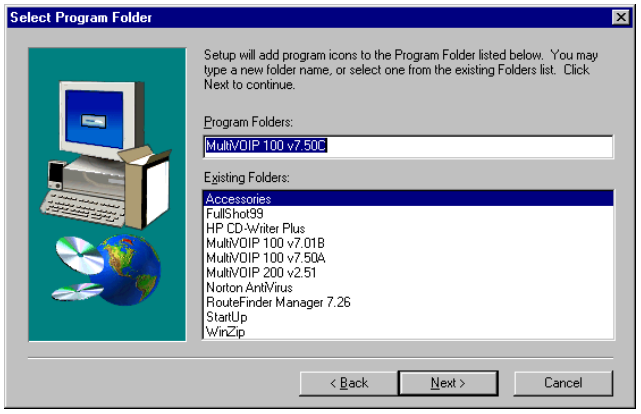
   Press **Enter** or click **Next** to continue.

6. The **Choose Destination Location** dialog box displays. Follow the on-screen instructions.


Choose Destination Location
Setup will install MultiVOIP 100 in the following folder.
To install to this folder, click Next.
To install to a different folder, click Browse and select another folder.
You can choose not to install MultiVOIP 100 by clicking Cancel to exit Setup.
Destination Folder
C:\VOIP7.50C
Browse...
< Back
Next >
Cancel

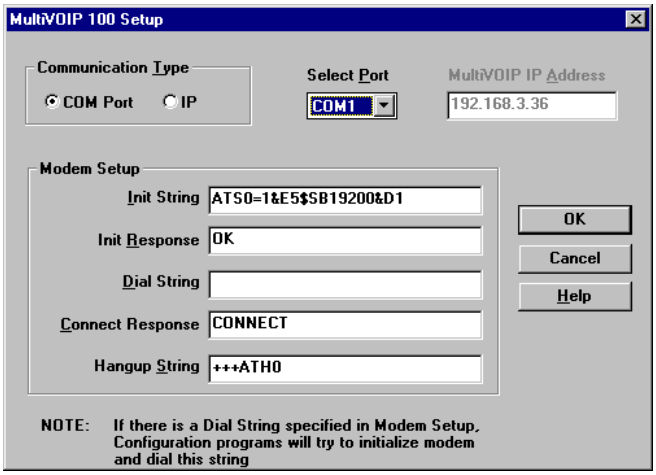
You can either choose the Destination Location of your MVP120 software or select the default destination by clicking **Next**. If you click **Browse,** you can select a different destination folder for the MVP120 software.

7. In the **Select Program Folder** dialog box, select where you want the program file to be located.



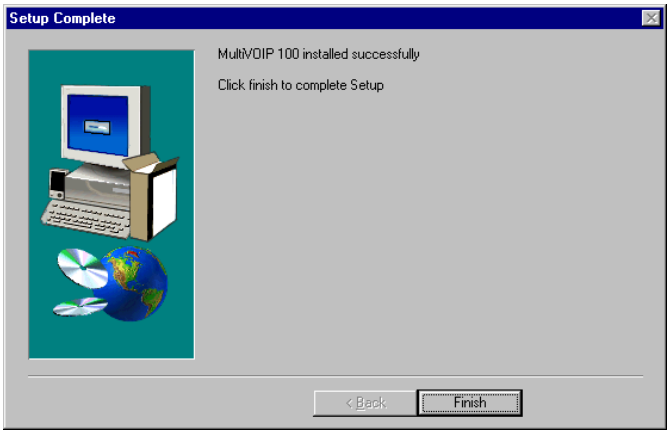
Verify the path and click **Next** to continue.

8. The **Copying program files window displays,** followed by the **MVP120 Setup** dialog box. This dialog box enables you to select the COM port of your PC that is connected to the Command port of the MVP120. From the **Select Port** list, select the COM port of your PC.

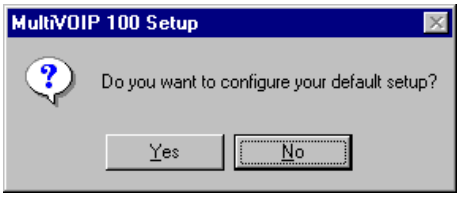


Click **OK** to continue.

9. The **Setup Complete** dialog displays.

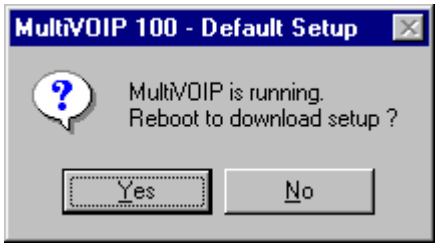


10. The following message displays:



Click **Yes** to continue.

11. The following message displays.



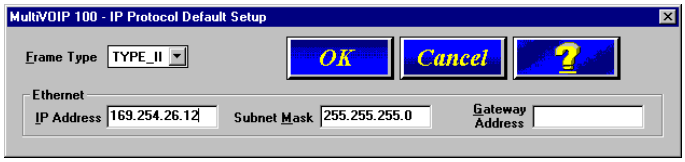
Click **Yes** to continue.

The **IP Protocol Default Setup** dialog box displays next (see page 16) as you continue configuring your MultiVOIP.

Configuring Your MultiVOIP

The following steps provide instructions for configuring your MVP120. The configuration sequence includes IP Protocol default setup, Channel setup, and Phone Directory Database setup. The Phone Directory Database setup is configured differently depending on whether or not the Gatekeeper function is available and enabled on the Phone Directory Database dialog box.

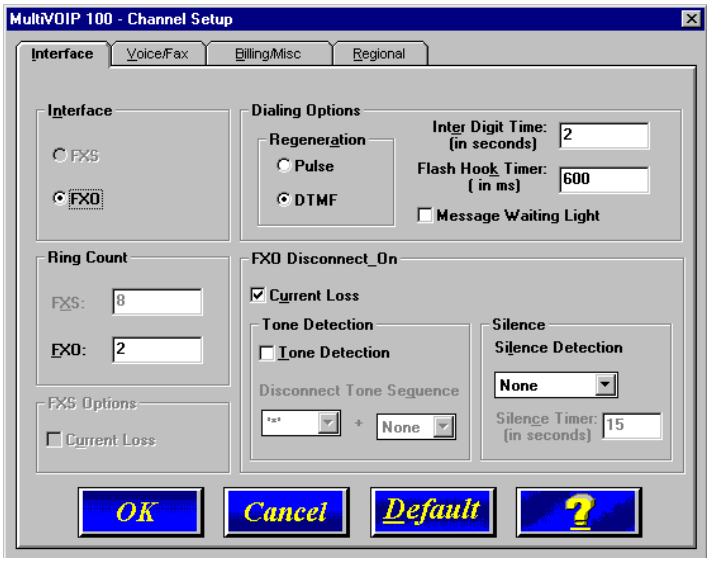
1. The **IP Protocol Default Setup** dialog box displays.



   The default **Frame Type** is **TYPE_II**. If this does not match your IP network, change the Frame Type by selecting **SNAP** from the **Frame Type** list. The available Frame Type choices are **TYPE_II** and **SNAP**.

2. In the **Ethernet** group, enter the **IP Address**, **Subnet Mask**, and **Gateway Address** unique to your IP LAN. The IP address is the unique LAN IP address assigned to the MVP120. The Gateway address is the IP address of the device connecting your MVP120 to the Internet. Click **OK** when you are finished.

3. The **Channel Setup** dialog box displays. The four tabs in this dialog box define the channel interface, voice/fax parameters, billing/miscellaneous parameters, and regional telephone parameters.



Check with your in-house phone personnel to verify whether your local PBX dial signaling is **Pulse** or **DTMF** (tone). Select the Regeneration option accordingly.

The **Inter Digit Time** option defines the maximum amount of time between dialed digits that the unit will wait before mapping the dialed digits to an entry in the Phone Directory Database. If too much time elapses between digits, the wrong number will be mapped and you will hear a rapid busy signal. If this happens, hang up and dial again. The default setting is **2** seconds.

In the **Flash Hook Timer** box, enter the amount of time for the duration of the flash hook signals output on the FXO interface. The default setting is 600 milliseconds.

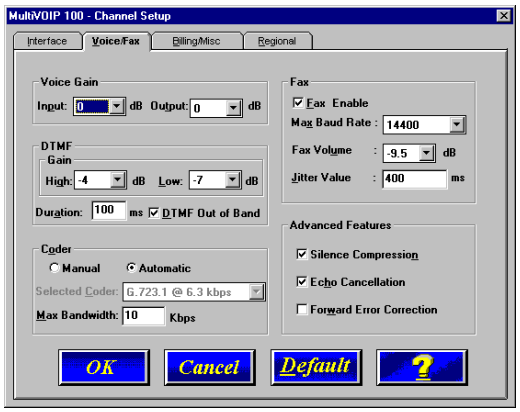
The **Message Waiting Light** check box must be selected on the originating and answering voice channel. This enables the number dialed to connect you to the appropriate voice channel, then output that number on the voice channel.

4. The **Ring Count FXO** box enables you to set the number of rings received on the FXO interface before answering the call. The default setting is **2** rings.

   Note: Zero (0) means no incoming calls will be answered.

   For FXO-to-FXO communications, you can enable a specific kind of FXO disconnect: **Current Loss**, **Tone Detection**, or **Silence Detection**. Check with your in-house phone personnel to verify the type of disconnect to use. If **Current Loss** is checked, the VOIP will hangup when it detects a loss of current on the FXO (phone) port. For tone detection, select from the lists one or two tones that will cause the line to disconnect. The person hanging up the call must then press the key or keys that produce those tones to hangup a call. For silence detection, select **One Way** or **Two Way**, then set the timer for the number of seconds of silence before disconnect. The default value of 15 seconds may be shorter than desired for your application.

5. The **Voice/Fax** tab displays the parameters for the voice gain, DTMF (Dual Tone Multi-Frequency) gain, voice coder, faxing, and advanced features such as Silence Compression, Echo Cancellation, and Forward Error Correction.



6. You can set up the input and output voice gain so that the volume can be increased or decreased. Input gain modifies the level of the audio coming into the voice channel before it is sent over the Internet to the remote MultiVOIP. Output gain modifies the level of the audio being output to the device attached to the voice channel. Make your selections from the Input and Output drop-down lists in the **Voice Gain** group. The valid range is +31dB to –31dB with a recommended/default value of **0**.

   You can also set up the DTMF gain (or output level in decibels - dB) for the higher and lower frequency groups of the DTMF tone pair. Make your selections in the drop-down lists in the **DTMF Gain** group. When **DTMF Out of Band** is checked, the unit reproduces the DTMF tones instead of passing them through.

   Note: Only change the DTMF gain under the direction of Multi-Tech Technical Support supervision.

7. To change the voice coder, select the **Manual** option in the Coder group. Select the appropriate coder from the **Selected Coder** list.

   If you changed the voice coder, ensure that the same voice coder is used on the voice/fax channel you are calling; otherwise, you will always get a busy signal.

   Note: If **Automatic Coder** is selected, enter the bandwidth in the **Max Bandwidth** box. Check with your Network Administrator to determine how much bandwidth is available.

8. The **Fax** group enables you to send/receive faxes on the voice/fax channel. You can set the maximum baud rate for faxes, the fax volume, and change the jitter value in milliseconds.

When receiving fax packets from a remote MultiVOIP, it is possible for individual packets to be delayed or received out of order due to traffic conditions on the network. To compensate for this effect, the MVP120 uses a Jitter Buffer. The **Jitter Value** box allows the MVP120 to wait a user-definable period of time, in milliseconds, for delayed or out of order fax packets. The range of allowable Jitter Values is 0 to 400 with a default of **400** milliseconds.

If you do not plan to send or receive faxes on the voice/fax channel, you can disable faxes in the Fax group.

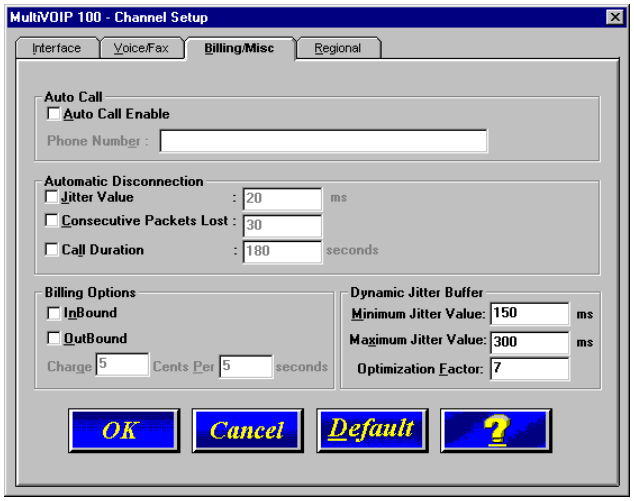
9. You can enable the voice/fax advanced features by clicking (checking) the silence compression, echo cancellation, or forward error correction options.

   The **Silence Compression** option defines whether silence compression is enabled for the voice channel. If silence compression is enabled, the MVP120 will not transmit voice packets when silence is detected, thereby reducing the amount of network bandwidth used by the voice channel.

   The **Echo Cancellation** option defines whether echo cancellation is enabled for the voice channel. If echo cancellation is enabled, the MVP120 will remove echo which improves the quality of sound.

   The **Forward Error Correction (FEC)** option defines whether forward error correction is selected for the voice channel. The FEC feature allows some of the voice packets that were corrupted or lost to be recovered. FEC adds an additional 50% overhead to the total network bandwidth consumed by the voice channel.

10. The **Billing/Misc** tab displays the parameters for auto call, automatic disconnection, billing options, and dynamic jitter buffer.



If you want to dedicate a local voice/fax channel to a remote voice/fax channel (so you will not have to dial the remote channel number), click the **Auto Call Enable** option in the **Auto Call** group. Then enter the phone number of the remote channel in the **Phone Number** box.

11. The **Automatic Disconnection** group provides three options to be used singly or in combination.

    The **Jitter Value** defines the average inter-arrival packet deviation (in milliseconds) before the call is automatically disconnected. Jitter is the inter-arrival packet deviation (phase shift of digital pulses) over the transmission medium that causes voice breakup which can be particularly disruptive to voice communications. The default setting is 20 milliseconds. A higher value means that the voice transmission will be more accepting of jitter. A lower value will be less tolerant of jitter.

    Consecutive Packets Lost defines the number of consecutive packets that are lost after which the call is automatically disconnected. The default setting is **30**.

Call Duration defines the maximum length of time (in seconds) that a call remains connected before the call is automatically disconnected. The default setting is **180** seconds. A call limit of three minutes may be too short for most configurations. Therefore, you may want to increase this default value.

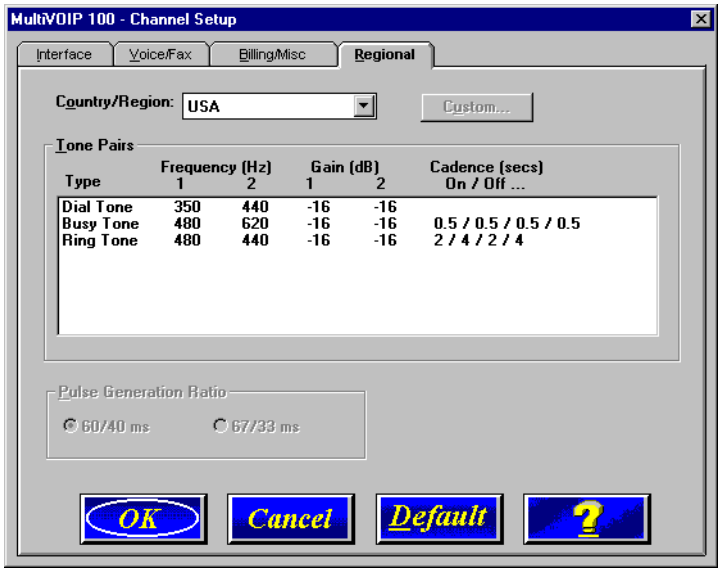
12. You can set billing options for inbound and/or outbound calls by checking them in the **Billing Options** group and then entering the charge in cents per number of seconds.

13. A minimum and maximum set of values can be set for **Dynamic Jitter Buffer**. When receiving voice packets from a remote MultiVOIP, it is possible to experience varying delays between packets due to traffic conditions on the network. This is called Jitter. To compensate for this effect, the MVP120 uses a Dynamic Jitter Buffer. The Jitter Buffer allows the MVP120 to wait for delayed voice packets by automatically adjusting the length of the Jitter Buffer between configurable minimum and maximum values. An **Optimization Factor** adjustment controls how quickly the length of the Jitter Buffer is increased when jitter increases on the network. The length of the jitter buffer directly effects the voice delay between MultiVOIP gateways.

    The Minimum Jitter Value default setting is **150** milliseconds, the Maximum Jitter Value default setting is **300** milliseconds, and the Optimization Factor default setting is **7**.

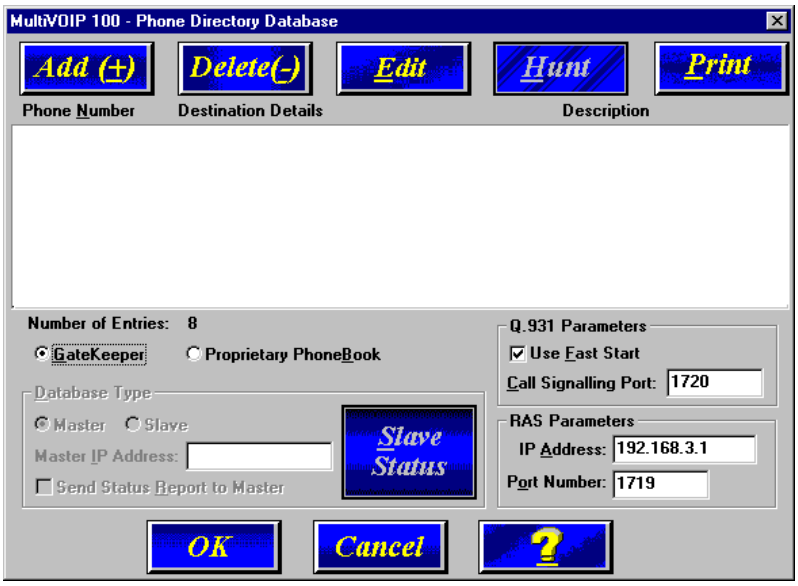
    If your country/region is not the default USA, click the **Regional** tab and proceed to step 14. Otherwise, click **OK** and proceed to step 15 to begin building your phone directory database.

14. To change the Tone Pairs on the **Regional** tab, select your country or region from the **Country/Region** list. If your country or region is not listed, click **Custom** to define it.



The **Tone Pairs** group enables you to select and modify the parameters. Click **OK** when finished.

15. The **Phone Directory Database** dialog box displays. This dialog box enables you to select either **GateKeeper** or **Proprietary Phone Book** as your type of phone book database. Once you have choosen the phone book database type, you can proceed to registering with a Gatekeeper in the following section or, if you are building a proprietary phone book database, proceed to **Building a Proprietary Phone Book Directory**.



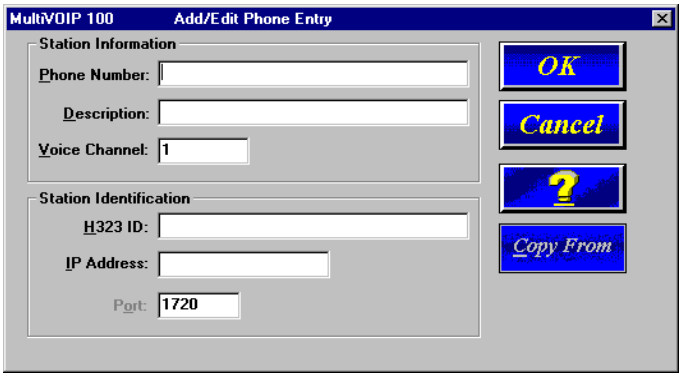
Registering with a Gatekeeper Phone Directory

This section describes how to register H.323 endpoints with the Gatekeeper. The H.323 Gatekeeper function resides at a PC acting as the central point for all calls within its zone and providing call control services to registered endpoints. The Gatekeeper performs two important call control functions: address translation from LAN aliases to IP addresses, and bandwidth management where the network manager has specified a threshold for the number of simultaneous conferences on the LAN.

In a GateKeeper environment, you will be enabling the GateKeeper option, entering an IP address for the GateKeeper, accepting the default port number, and if the GateKeeper network is servicing Fast Start, accept the defaults in the Q.931 Parameters group. However, if this network zone is primarily non-fast start supported, clear the **Use Fast Start** check box.

1. Enable the **Gatekeeper** option
2. If the GateKeeper network employs Fast Start, then accept the **Use Fast Start** option (default) in the **Q.931 Parameters** group. You may have to verify this with the GateKeeper administrator. If **Use Fast Start** option is selected, accept the **Call Signaling Port** default **1720**.
3. Enter the Gatekeeper IP Address in the **IP Address** box of the **RAS Parameters** group.
4. Accept the default **Port Number 1719**.

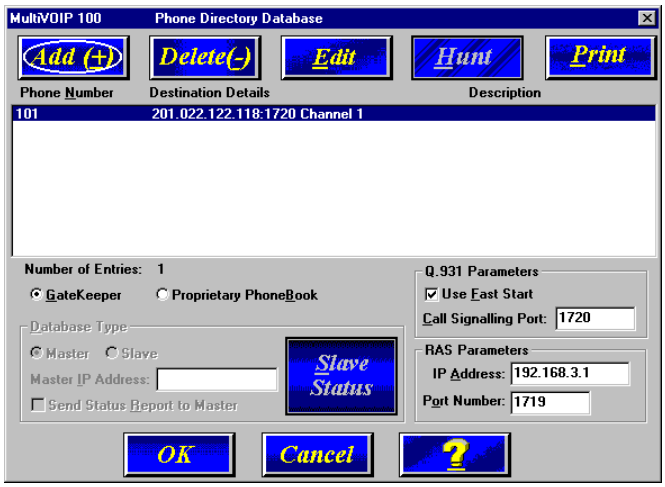
   CAUTION: The default setting for the Gatekeeper Port Number is **1719**. This can be changed to a different value by the Gatekeeper administrator. If you decide to change the default Port Number, you must use the same number on the Gatekeeper and all other H.323 endpoints.
5. When you are finished with this dialog box, click **Add** to begin building your phone directory database. The **Add/ Edit Phone Entry** dialog box displays.



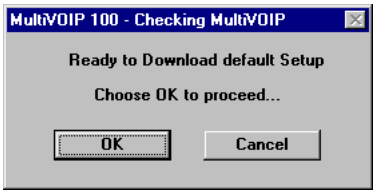
6. Enter the unique phone number of the local device in the **Phone Number** box.
7. Leave the **Description** box blank.

8. Enter the **Voice Channel** number corresponding to the phone number entered. The MVP120 will always be **1**.
9. Fill in the **H.323 ID** box with a description to identify the phone number. For this example, you could enter “New York Office.”
10. Enter the IP Address of the MultiVOIP you are currently configuring in the **IP Address** box.

11. Click **OK** when you are finished and the **Phone Directory Database** dialog box displays with your first entry.



12. Click **OK** when you are finished.
13. Click **OK** to download setup.
14. The following dialog box displays.



Click **OK** to download default setup.

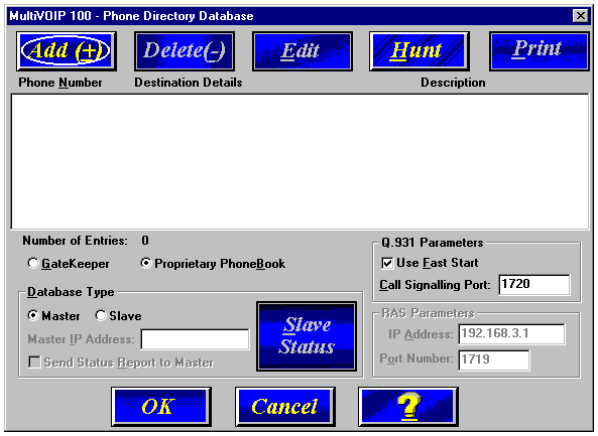
15. After the setup has been written to the MVP120, the unit is rebooted.
16. Verify that the **BOOT** LED on the MVP120 is off after the download completes. This may take several minutes as the MVP120 reboots.

    Note: Since the MVP120 doesn't have a power LED, no LEDs will be lit if there is no activity after boot up, unless the ethernet port is connected to the network.

Building a Proprietary Phonebook Directory

1. To build your proprietary Phone Directory (in an H.323 environment without the Gatekeeper option enabled), you will first need to select the **Proprietary Phonebook** option and then configure the “Master” MVP120 and then add the “Slave” MultiVOIPs (or other H.323 endpoints). Configuring the “Slave” MultiVOIPs is discussed later.

   The MultiVOIP, configured as **Master**, contains the proprietary phonebook database. All other MultiVOIPs added to the proprietary phonebook database are designated **Slaves**. The Master database contains the phone numbers of all H.323 endpoints available for communication on an IP network. This database is downloaded to each Slave MultiVOIP as it comes online.

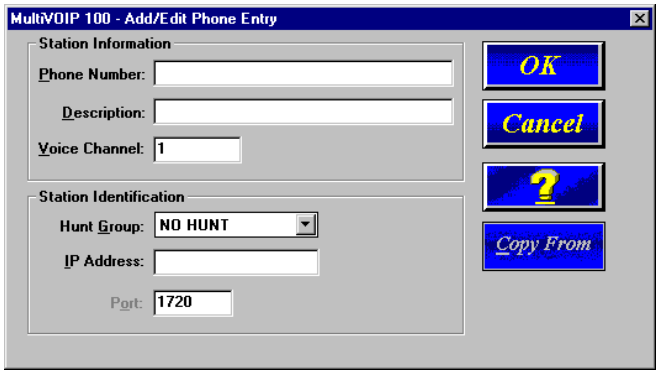


2. To configure the Master MVP120, make certain that the **Proprietary Phonebook** and **Master** options are enabled. The **Master IP Address**, **Send Status Report to Master**, and **RAS Parameters** group will be disabled.

   The **Slave Status** button displays the **Slave VOIP Status** dialog box used for viewing phone number, IP address,

status, and description of slave units (See “Configuring Your Slave MVP120s” for details). Note: In the Q.931 Parameters group, **Use Fast Start** is checked for compatibility with other H.323 devices that support Fast Start capability.

Click **Add** to begin building your phone directory database. The **Add/Edit Phone Entry** dialog box displays.



Configuring Your Master MVP120

1. Enter a unique phone number for the local VOIP in the **Phone Number** box, and enter **1** for the voice/fax channel number in the **Voice Channel** box. The number must be **1** for each MVP120.

2. The **Description** box is optional, but can be useful in associating the channel with the location. If you wish, enter a description of your local phone number. This description serves to identify the phone number you entered in the previous step. Normally the Master MultiVOIP resides at the entity’s main office. Therefore, for this example you could enter a description such as “New York Office”.

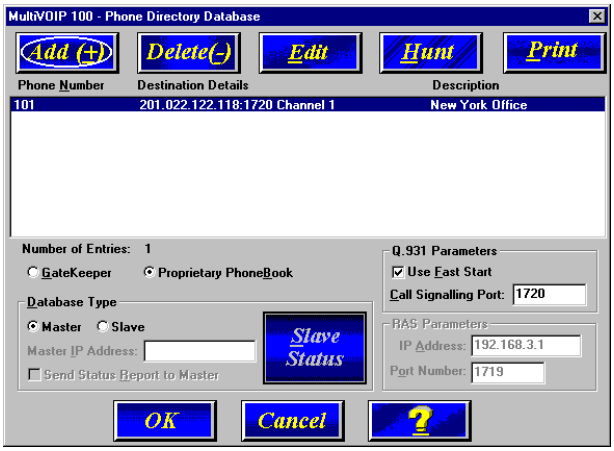
3. The **Station Identification** group includes a **Hunt Group** list. This list enables you to indicate which Hunt Group you want the phone number to be associated with. Select the Hunt Group number from the Hunt Group drop down list box, or you can select NO HUNT if you do not want this entry to participate in hunting.

   Note: Hunting is a series of telephone lines organized in such a way that if the first line is busy the next line is called or hunted until a free line is found.

   Once you have selected a Hunt Group (or NO HUNT), you must enter the IP Address of the Master MVP120 in the **IP Address** box.

   Note: The **Port** box becomes active as you begin to enter the IP Address. The entry is the H.323 industry standard Port value (1720) used to communicate with other H.323 endpoints.

4. Click **OK** to return to the **Phone Directory Database** dialog box. It now includes the phone number, destination details, and description of Hunt Group #1.



You have completed configuration of the Master MVP120. The voice channel you have set up is labeled Hunt Group #1. If a call from an H.323 endpoint (a MultiVOIP or a stand-alone H.323 endpoint) to Phone Number 101 is unable to be connected, it will automatically connect to the next available phone number in Hunt Group #1, such as **Phone Number** on a Slave MultiVOIP.

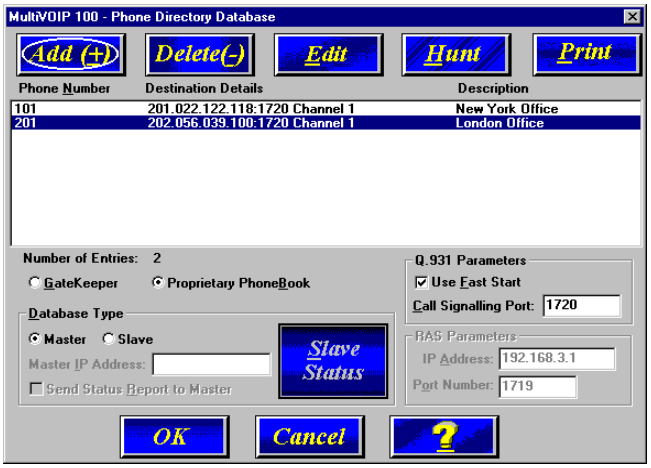
5. Add all other phone numbers (Slave units and stand-alone units) to the Phone Directory database. To add a Channel of a Slave MultiVOIP, click **Add.** The **Add/Edit Phone Entry** dialog box displays again.

6. Enter the phone number for the Slave MultiVOIP in the **Phone Number** box.

7. Click in the **Description** box. Enter a remote MultiVOIP phone number description for the Voice/Fax Channel.

8. In the **Station Identification** group, select **Hunt Group #2** from the **Hunt Group** list, enter the London Office's IP Address (202.056.039.100), and accept the H.323 industry standard Port value (1720) used to communicate with other H.323 endpoints.

9. Click **OK** and you are returned to the **Phone Directory Database** dialog box which now includes the remote phone number and related information in the **Phone Number** list.



10. To configure a stand-alone endpoint (for example, a PC with NetMeeting software), click **Add** and the **Add/Edit Phone Entry** dialog box displays again.

11. Enter the phone number for the stand-alone endpoint in the **Station Information** group **Phone Number** box. For example, 301.

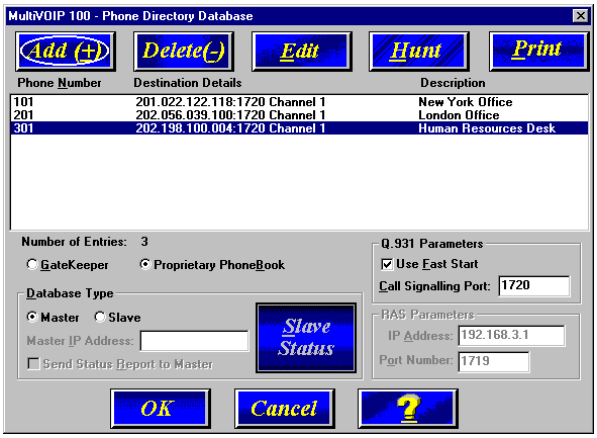
12. Click inside the **Description** box and enter a description for the remote NetMeeting. For example, “Human Resources Desk”.

    Note: Because the H.323 endpoint is not a MVP120, the Phone Directory database ignores the Voice Channel entry. It does not matter what value is entered.

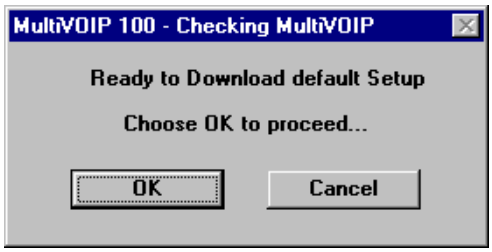
13. In the **Station Identification** group, select NO HUNT from the **Hunt Group** list, enter the Human Resource Desk’s IP Address (202.198.100.04) and accept the H.323 industry standard Port value (1720) used to communicate with other H.323 endpoints.

    Note: This stand-alone was not configured as part of a Hunt Group. However, depending on your requirements, you could configure a stand-alone to be part of a Hunt Group.

14. Click **OK** and you are returned to the **Phone Directory Database** dialog box which now includes the stand-alone phone number and related information in the **Phone Number** list.



15. When you have finished, click **OK** to download the default setup.



16. Click **OK**. After the setup is written to the MVP120, the unit reboots.

17. Check to ensure that the **BOOT** LED on the MVP120 is off after the download is complete. This may take several minutes as the MVP120 reboots.

    At this time your master MVP120 is configured. Proceed to the “Configuring Your Slave MVP120s” section.

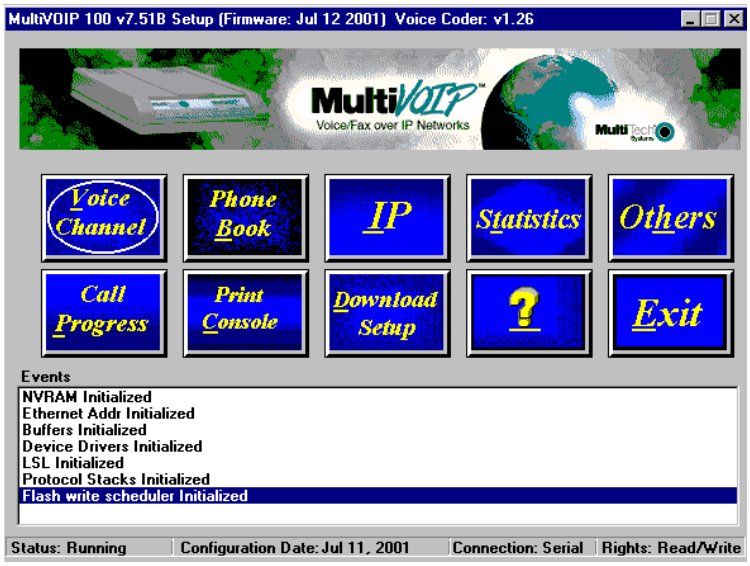
Configuring Your Slave MVP120s

If the **Proprietary Phonebook** option on the **Phone Directory Database** dialog box was enabled, then you will need to configure all remote H.323 endpoints as “Slave” units. For example, the MVP120 at the company’s subsidiary office in London would need to be configured as a “Slave”.

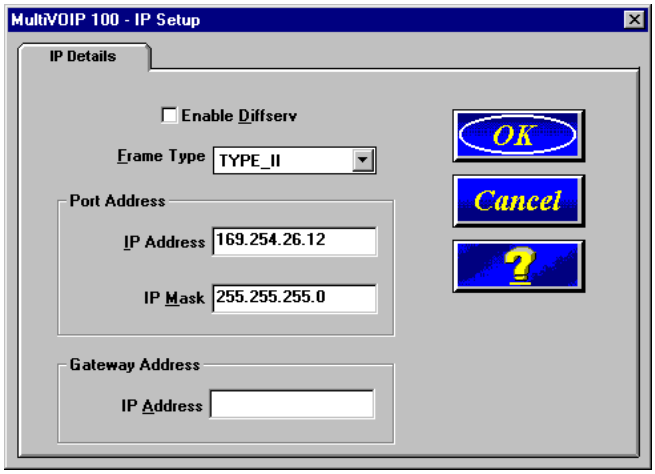
CAUTION: If you are installing a MVP120 behind a Firewall, the Firewall must support H.323. Refer to your Firewall user documentation to enable H.323 support.

1. Disconnect the PC from the Command port of the Master MVP120 and connect it to the Command port on the Slave MVP120.

2. From your desktop, click **Programs I MultiVOIP 100 I MultiVOIP 100 Configuration**. The Main menu displays.



3. Click **IP** to display the **IP Setup** dialog box.



Select the **Enable Diffserv** check box if you have routers that support Diffserv (sometimes called IP Precedence). This feature gives priority to voice packets so they are not delayed because of large data files being downloaded.

The default **Frame Type** is **TYPE_II**. If this does not match your IP network, select the Frame Type from the Frame Type list. The Frame Type choices are **TYPE_II** and **SNAP**.

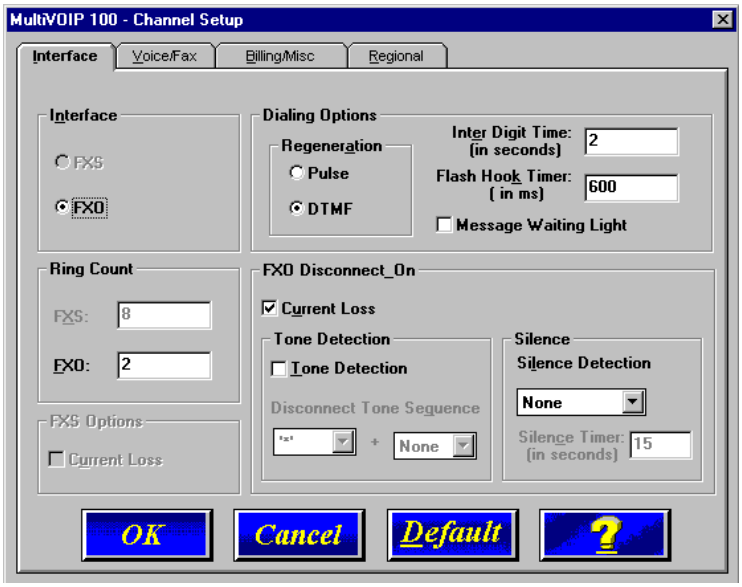
4. In the **Port Address** group, enter the **IP Address** and **IP Mask**. In the **Gateway Address** group, enter the gateway IP address for the slave unit.

   The IP Address is the unique IP address that you assign to the MVP120, and the Gateway Address is the IP address of the device (such as a network router) connected to the Internet/Intranet.

   Click **OK** when you are finished. The Main menu displays.

5. From the Main menu, click **Voice Channel** to display the **Channel Setup** dialog box. The **Channel Setup** dialog box displays. The four tabs in this dialog box define the

channel interface, voice/fax parameters, Billing/Misc parameters, and regional telephone parameters for the channel.



Feature options are enabled or disabled according to the interface type. The **Inter Digit Time** option defines the maximum amount of time that the unit will wait before mapping the dialed digits to an entry in the Phone Directory Database. If too much time elapses between digits, and the wrong numbers are mapped, you will hear a rapid busy signal. If this happens, it will be necessary to hang up and dial again. The default is 2 **seconds.**

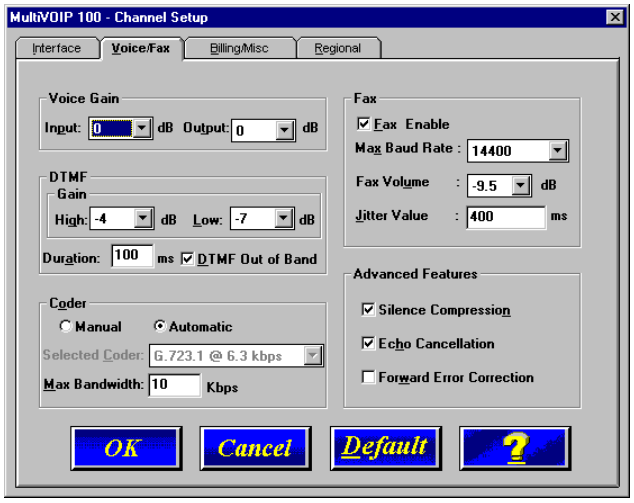
6. The **Ring Count FXO** box enables you to set the number of rings received before the FXO interface answers the call. The default setting is **2** rings.

   Note: Zero (0) means incoming calls will not be answered.

The **Message Waiting Light** check box must be selected on the originating and answering voice channel. This enables the number dialed to connect you to the appropriate voice channel, then output that number on the voice channel.

For FXO-to-FXO communications, you can enable a specific kind of FXO disconnect: **Current Loss**, **Tone Detection**, or **Silence Detection**. Check with your in-house phone personnel to verify the type of disconnect to use. If **Current Loss** is checked, the VOIP will hangup when it detects a loss of current on the FXO (phone) port. For tone detection, select from the lists one or two tones that will cause the line to disconnect. The person hanging up the call must then press the key or keys that produce those tones to hangup a call. For silence detection, select **One Way** or **Two Way**, then set the timer for the number of seconds of silence before disconnect. The default value of 15 seconds may be shorter than desired for your application.

7. The **Voice/Fax** tab displays the parameters for the voice gain, DTMF (Dual Tone Multi-Frequency) gain, voice coder, faxing, and advanced features such as Silence Compression, Echo Cancellation, and Forward Error Correction.



8. Input gain modifies the level of the audio coming in to the voice channel before it is sent over the Internet to the remote MultiVOIP. Output gain modifies the level of the audio being output to the device attached to the voice channel. Make your selections from the Input and Output lists in the **Voice Gain** group. The valid range is +31dB to –31dB with a recommended/default value of **0**.

   You can also set up the DTMF gain (or output level in decibels - dB) for the higher and lower frequency groups of the DTMF tone pair. Make your selections in the drop-down lists in the **DTMF Gain** group.

   Note: Only change the DTMF gain under the direction of Multi-Tech Technical Support supervision.

9. To change the voice coder, select the **Manual** option in the Coder group. Select the appropriate coder from the **Selected Coder** list.

   If you changed the voice coder, ensure that the same voice coder is used on the voice/fax channel you are calling; otherwise, you will always get a busy signal.

Note: If the **Automatic Coder** is selected, enter the bandwidth in the **Max Bandwidth** from the list. Check with your VOIP administrator to determine how much bandwidth is available.

10. The **Fax** group enables you to send/receive faxes. You can set the maximum baud rate for faxes and the fax volume in the two lists and change the jitter value in milliseconds.

    When receiving fax packets from a remote MultiVOIP, it is possible for individual packets to be delayed or received out of order due to traffic conditions on the network. To compensate for this effect, the MVP120 uses a Jitter Buffer. The **Jitter Value** box allows the MVP120 to wait a user-definable period of time, in milliseconds, for delayed or out of order fax packets. The range of allowable Jitter Values is 0 to 400 with a default of **400** milliseconds.

    If you do not plan to send or receive faxes on a voice/fax channel, you can disable faxes in the Fax group.

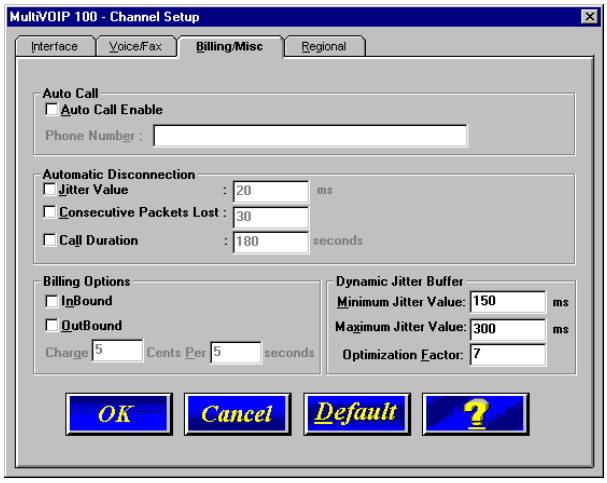
11. You can enable the voice/fax advanced features by clicking (checking) the silence compression, echo cancellation, or forward error correction options.

    The **Silence Compression** option defines whether silence compression is selected. If silence compression is enabled, the MVP120 will not transmit voice packets when silence is detected, thereby reducing the amount of network bandwidth that is being used by the voice channel.

    The **Echo Cancellation** option defines whether echo cancellation is enabled for the Voice/Fax Channel. If echo cancellation is enabled, the MVP120 will remove echo-delay which improves the quality of sound.

The **Forward Error Correction (FEC)** option defines whether forward error correction is enabled for the voice channel. The FEC feature allows some of the voice packets that were corrupted or lost to be recovered. FEC adds an additional 50% overhead to the total network bandwidth consumed by the voice channel.

The **Billing/Misc** tab displays the parameters for auto call, automatic disconnection, billing options and dynamic jitter buffer.



12. If you want to dedicate a local voice/fax channel to a remote voice/fax channel (so you will not have to dial the remote channel), click the **Auto Call Enable** option in the **Auto Call** group. Then enter the phone number of the remote MultiVOIP in the **Phone Number** box.

13. The **Automatic Disconnection** group provides three options to be used singly or in combination.

The **Jitter Value** defines the average inter-arrival packet deviation (in milliseconds) before the call is automatically disconnected. Jitter is the inter-arrival packet deviation (phase shift of digital pulses) over the transmission medium that causes voice breakup which can be particularly disruptive to voice communications. The default setting is 20 milliseconds. A higher value means that the voice transmission will be more accepting of jitter. A lower value will be less tolerant of jitter.

Consecutive Packets Lost defines the number of consecutive packets that are lost after which the call is automatically disconnected. The default is **30** packets.

Call Duration defines the maximum length of time (in seconds) that a call remains connected before the call is automatically disconnected. The default setting is **180** seconds. A call limit of three minutes may be too short for most configurations. Therefore, you may want to increase this default value.

14. You can set billing options for inbound and/or outbound calls by checking them in the **Billing Options** group and then entering the charge in cents per number of seconds.

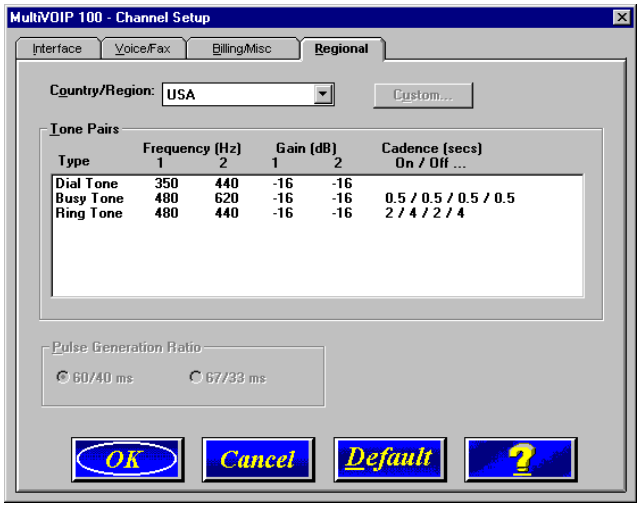
15. A minimum and maximum set of values can be set for **Dynamic Jitter Buffer**. When receiving voice packets from a remote MultiVOIP, it is possible to experience varying delays between packets due to traffic conditions on the network. This is called Jitter. To compensate for this effect, the MVP120 uses a Dynamic Jitter Buffer. The Jitter Buffer allows the MVP120 to wait for delayed voice packets by automatically adjusting the length of the Jitter Buffer between configurable minimum and maximum values. An **Optimization Factor** adjustment controls how quickly the length of the Jitter Buffer is increased when jitter increases on the network. The length of the jitter buffer directly effects the voice delay between MVP120 gateways.

The Minimum Jitter Value default setting is **150** milliseconds, the Maximum Jitter Value default setting is **300** milliseconds, and the Optimization Factor default setting is **7**.

If your country/region is not the default USA, click the **Regional** tab and proceed to step 17.

16. To change the Tone Pairs on the **Regional** tab, click the **Country/Region** down arrow and highlight your specific country or region.

    Note: If your country or region is not listed, click **Custom** to define it.

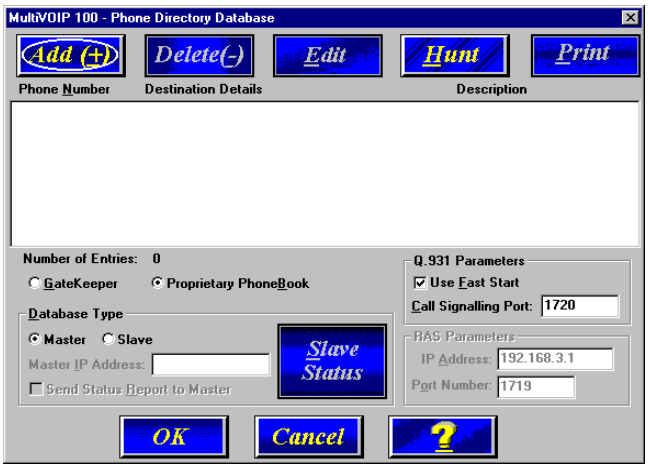


    The **Tone Pairs** group enables you to select/modify the parameters. Click **OK** when finished.

17. From the Main menu, click **Phone Book** to display the **Phone Directory Database** dialog box.

    Note: Refer to pages 24-27 if you had chosen Gatekeeper as the type of phone book database during the master phonebook setup.

Make certain the **Proprietary Phonebook** option is enabled and in the **Database Type** group, select the **Slave** option. The **Master IP Address** box becomes active.



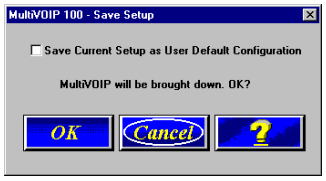
Note: After you have enabled the **Slave** option, the **Slave Status** button is replaced by the **Update** button. Once your Phone Directory database has been created, you can click this button to refresh the entries in the Phone Directory Database window.

18. Enter the IP address (204.022.122.118) of the New York Office MVP120 in the **Master IP Address** box and enable the **Send Status Report to Master** so that status reports are sent to the Master MVP120.

    Note: In a Dial-On-Demand (DOD) network, you should leave **Send Status Report to Master** disabled. This allows the router to disconnect whenever there is no voice activity. Note that Slaves with **Send Status Report to Master** disabled will show up as “Unknown” when viewing Slave status on the Master.

19. Click **OK** to return to the Main menu.

20. Click **Download Setup** to write the new configuration to the slave unit. The **Save Setup** dialog box displays.



21. Select the Save Current Setup as User Default Configuration check box and click **OK**. The setup configuration is written to the MVP120. After the setup is written to the MVP120, it reboots.
22. Check that the **Boot** LED on the MVP120 is off after the download is complete. This may take several minutes as the MVP120 reboots.
23. You are returned to the Main menu.

Your MVP120 is operational at this time.

Repeat the process for each of the slave units. When all slaves have been configured, go to “Deploying the VOIP Network.”

Deploying the VOIP Network

For a Proprietary Phone Directory database, the VOIP administrator can deploy the pre-configured slave MVP120s to their remote sites. The remote site administrators need only connect power to the pre-configured MVP120, connect the MVP120 to their Ethernet LAN and predefined telephone equipment, and then wait for the phone directory database to be downloaded.

With the **Gatekeeper** option enabled on the **Phone Directory Database** dialog box, all MVP120s are configured as “Gatekeeper” and cannot be downloaded. In this case, each MVP120 Phone Book will be programmed with phone numbers for its own channel. This phone number is registered with the H.323 Gatekeeper (See the “Registering with a Gatekeeper Phone Directory” section discussed earlier.)

Remote Site Administrator

The following steps are for MVP120 H.323 endpoints. For non-MVP120 H.323 endpoints, refer to the appropriate installation documentation.

1. Connect one end of the power supply to a live AC outlet and connect the other end to the **Power** connection on your MVP120 (See Figure 4).

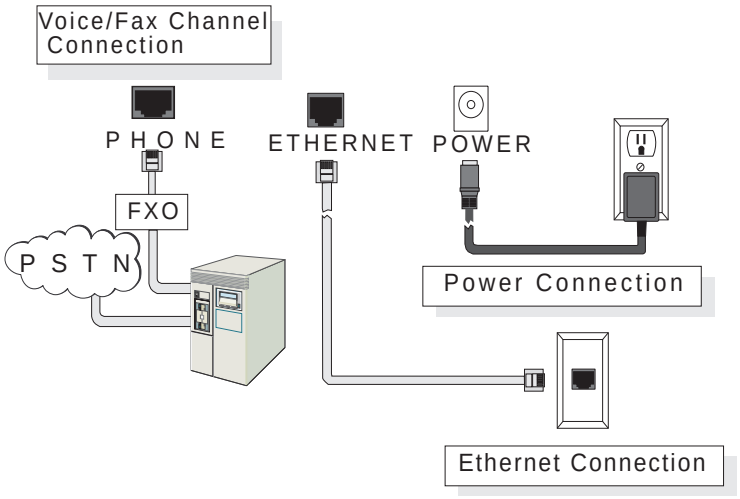


Figure 4. Remote Site Cable Connection

2. Connect a network cable to the **Ethernet (**RJ-45) connector on the back of your MVP120.

3. The phone port is used to connect to an FXO device. Connect an RJ-11 cable from the pnone connector to a PBX analog port or a phone line from the phone company.
4. Turn on power to the MVP120 by placing the ON/OFF switch on the right side panel to the ON position. Wait for the BOOT LED on the MVP120 to go off before proceeding. This may take a couple of minutes.
5. At this time your VOIP network should be fully operational. Dial one of the sites in your network.

Limited Warranty

Multi-Tech Systems, Inc. (“MTS”) warrants that its products will be free from defects in material or workmanship for a period of two years from the date of purchase, or if proof of purchase is not provided, two years from date of shipment. MTS MAKES NO OTHER WARRANTY, EXPRESSED OR IMPLIED, AND ALL IMPLIED WARRANTIES OF MERCHANTABILITY AND FITNESS FOR A PARTICULAR PURPOSE ARE HEREBY DISCLAIMED. This warranty does not apply to any products which have been damaged by lightning storms, water, or power surges or which have been neglected, altered, abused, used for a purpose other than the one for which they were manufactured, repaired by the customer or any party without MTS’s written authorization, or used in any manner inconsistent with MTS’s instructions.

MTS’s entire obligation under this warranty shall be limited (at MTS’s option) to repair or replacement of any products which prove to be defective within the warranty period, or, at MTS’s option, issuance of a refund of the purchase price. Defective products must be returned by Customer to MTS’s factory transportation prepaid.

MTS WILL NOT BE LIABLE FOR CONSEQUENTIAL DAMAGES AND UNDER NO CIRCUMSTANCES WILL ITS LIABILITY EXCEED THE PURCHASE PRICE FOR DEFECTIVE PRODUCTS.

FCC Declaration

NOTE: This equipment has been tested and found to comply with the limits for a **Class A** digital device, pursuant to Part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference when the equipment is operated in a commercial environment. This equipment generates, uses and can radiate radio frequency energy, and if not installed and used in accordance with the instructions, may cause harmful interference to radio communications. Operation of this equipment in a residential area is likely to cause harmful interference in which case the user will be required to correct the interference at his own expense.

Warning: Changes or modifications to this unit not expressly approved by the party responsible for compliance could void the user’s authority to operate the equipment.

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